What is claimed is:

1. A method of compensating within a receiving endpoint for lost audio packets transmitted across an IP network, comprising the steps of:

storing a packet buffer of samples as a plurality of sub packets within a buffer;

inserting at least one interpolated sub packet between successive sub packets in said buffer; and

playing out said sub packets from said buffer.

2. The method of claim 1, wherein each said interpolated sub packet comprises a weighted average of present and next ones of said successive sub packets to be played out of said buffer such that first samples of the interpolated sub packet resemble first samples of said next one of said successive sub packets.

3. The method of claim 2, wherein said weighted average is: $PN_{m=0-(M-1)} = (mP_m + (M-m)N_m)/M$, wherein $P_{0-(M-1)}$ represents samples 0 to (M-1) of M samples of the present sub packet, and $N_{0-(M-1)}$ represents samples 0 to (M-1) of M samples of the next sub packet.

4. The method of claim 3, wherein each said interpolated sub packet is inserted as follows:

```
#define SUB_PACKET_SIZE     8               /* In samples */
int current_sp[ SUB_PACKET_SIZE ];
int nex_sp[ SUB_PACKET_SIZE ];

void smooth_sub_packet( void )
{
   current_sp[0] = (current_sp[0] + 3 * next_sp[0]) >> 2;
   current_sp[1] = (2 * current_sp[1] + 2 * next_sp[1]) >> 2;
   current_sp[2] = (3 * current_sp[2] ÷ next_sp[2]) >> 2;
}

If ( packet compensation mode AND in_odd_sub_packet )
{
   CALL smooth_sub_packet and RESEND current_sp
}
```

5. The method of claim 1, further comprising the step of inserting interpolated sub packets between every other one of said sub packets in said buffer.

6. The method of claim 5, wherein each of said sub packets is of 1 ms duration.

7. The method of claim 1, wherein said step of inserting at least one interpolated sub packet between successive sub packets is only performed when said buffer contains less than a predetermined threshold number of sub packets.

8. The method of claim 7, wherein said predetermined threshold number of sub packets is equivalent to the number of samples in a single packet buffer.